Date: Wed, 4 Aug 93 04:30:28 PDT
From: Ham-Homebrew Mailing List and Newsgroup <ham-homebrew@ucsd.edu>
Errors-To: Ham-Homebrew-Errors@UCSD.Edu
Reply-To: Ham-Homebrew@UCSD.Edu
Precedence: Bulk
Subject: Ham-Homebrew Digest V93 #1
To: Ham-Homebrew

Ham-Homebrew Digest   Wed, 4 Aug 93   Volume 93 : Issue   1

Today's Topics:

Date: Wed, 4 Aug 1993 00:39:20 GMT
From: elroy.jpl.nasa.gov!usc!howland.reston.ans.net!torn!mcshub!nimios!kirkland@decwrl.dec.com
Subject: Acquiring Crystal Filters
To: ham-homebrew@ucsd.edu

Does anyone out know of suppliers for crystal filters (i.e. single quantities)
for 9 - 10 MHz IF and 40 MHz IF or higher?

Thanks Bill Kirkland

reply to kirkland@comres3.Eng.McMaster.CA

Date: 3 Aug 1993 14:43:35 GMT
From: news.graphics.cornell.edu!newsstand.cit.cornell.edu!newsstand.cit.cornell.edu!usenet@tcgould.tn.cornell.edu

Subject: Audio filter for headphones
To: ham-homebrew@ucsd.edu

In article <104090001@hpl-opus.hpl.hp.com> Rick Walker,
walker@hpl-opus.hpl.hp.com writes:
>This would work great over stereo loudspeakers, but would not be quite
>right over headphones. For playback over speakers *amplitude*
differences
>turn into a phase difference due to phase cancellation from the right
>and left speakers at your ears. For headphone reproduction, this
>cancellation is lacking so you need to synthesize a *phase*
>difference as a function of frequency to give a spatial variation to
>the tones.

Hmmm - this might be true, but I don't remember the article doing that.
What I remember was just a pair of mirror image filters around the center
frequency - as I described. And I'm pretty sure the author used them with
headphones - not speakers.

>
>A suitable circuit would be a good, mono filter driving your right ear
and
>followed by an allpass network to drive your left ear. The phase delay
>between the channels will be interpreted by your brain as a difference in
>location.
>
>--
>Rick Walker
>wb6gvi

So how do you get the signals to change from left to right as the tone
changes. I can see how you could get a fixed phase difference with the
allpass network, but what the system wants is a change in relative signal
to left and right ear dependent on frequency. Maybe we are working at two
different things here.

Anybody remember what magazine the article was in? We've got a fair
collection for QST and 73 in the university libraries - maybe I can go
spelunking at lunch. Even a rough idea of the year (or decade) would
help. I'm thinking late 70's maybe.

73 de Kevin, WB2EMS (fkf1@cornell.edu)

Date: Wed, 4 Aug 1993 01:51:45 GMT
From: usc!sdd.hp.com!col.hp.com!news.dtc.hp.com!hpscit.sc.hp.com!hplextra!hpl-
opus!walker@network.ucsd.edu

Subject: Audio filter for headphones
To: ham-homebrew@ucsd.edu

In rec.radio.amateur.homebrew, F. Kevin Feeney <fkf1@cornell.edu> writes:

```
> So how do you get the signals to change from left to right as the tone
> changes. I can see how you could get a fixed phase difference with the
> allpass network, but what the system wants is a change in relative signal
                                                ^^^^^^^^^^^^^^^^^^^^^^^^^^^
> to left and right ear dependent on frequency. Maybe we are working at two
> different things here.
```

To simulate what the ear hears when you move a source from right to left
you don't want to make a relative signal strength change. You need to
make a phase change. This confusion comes from the fact that to make
a phase change with stereo speakers, you need to make an amplitude change
between the speakers. This is *not* true for headphone reproduction.

To see why this is, you can analyze what the ear hears in the following
experiment:

```
    (*) sound source
     |\
     | \
     |  \
     |   \
     |    \
    (L)   (R)         Right ear signal is delayed from Left ear signal

       (*) sound source
       /\
      /  \
     /    \
    (L)   (R)         No phase lag between signals at each ear
```

Notice that the amplitude stays constant and the phase varies when
the source is moved from left to right. Also, at any given position,
their is a fixed *time delay* difference between the ears giving a
phase shift which is frequency dependant.

One way to achieve this is to simulate the paths in a stereo speaker
system. (If you draw out the paths from the L/R speakers to the ears,
you can verify that amplitude changes in the stereo speakers will convert
to the proper *phase* variations at each ear - for details, look up the
original patent on stereo by Blumlein)

```
                          (Mono Source)
```

```
                                |
                                |
                        (500Hz - 1KHZ BPF)
                                |
                    -------------------------
                    |                       |
              ( 750 Hz HPF)            ( 750 Hz LPF)
                    |                       |
                    +->------       -----<--+
                    |        \     /        |
               (1 ms delay)   \   /    (1 ms delay)
                    |          \ /          |
                    |           \           |
                    |          / \          |
                    |         /   \         |
                    |        /     \        |
                 ( SUM )-<---       --->-( SUM )
                    |                       |
                    |                       |
                    V                       V

                To left ear            To right ear
```

You can achieve the 1 ms delay with op-amp all-pass sections, or with Reticon bucket brigade delay lines. "1ms" is chosen to approximate the time delay between ears.

I've built this network (without the HP and LPFs) so that I could listen to conventional stereo and have it sound right over headphones. It's fun to dial the delay and have the apparent acoustic location of the speakers move around.

I think you need to do something equivalent to the above if you want to really make a stereo sound stage with the low frequencies on the left and the high frequencies on the right. It may be that the original simple technique approximates the above "correct" behavior over a small band about the filter crossover frequency. With a narrow passband, this may be good enough.

A really "fancy" version of this circuit would include filters in the direct and crossfeed paths to simulate the variation in frequency response of the ear as a function of angle.

--
Rick Walker
WB6GVI

Date: 3 Aug 93 14:55:09 GMT
From: uplherc!wpsun4!mb@uunet.uu.net
Subject: Direct Digital Synthesis in Aug 93 issue of 73
To: ham-homebrew@ucsd.edu

linnig@m2000.dseg.ti.com (Mike Linnig) writes:
:
: My question is, does anyone know where I can buy a HSP45102 chip?   What is
: the approximate cost?

I ordered the HSP45102 and accompanying DAC from Wyle last week.   As I recall, the DDS chip was about $25, and the DAC was just under $20. I ordered two of each to be able to get quadrature output for an R2/T2 phasing SSB receiver/ transmitter that I'll be building next. I haven't found a source (besides the author) for the Coilcraft filter mentioned in the article.

Michael Bendio (mb@titan.wordperfect.com)

Date: 3 Aug 1993 11:33:52 GMT
From: psgrain!ee.und.ac.za!shrike.und.ac.za!pc-bdonal.ee.und.ac.za!
bdonal@uunet.uu.net
Subject: Looking for a synt chip for a 10M HT.
To: ham-homebrew@ucsd.edu

Hi,

I am trying to find a synthesiser chip that will work at 10M and below, so that I can upgrade my crystal tuned rig. I guess there are a number of parts that will fit the bill, but which is most common/easy to use.

Any pointers would be great.

Thanks, Brian

```
             _____       _____
Brian.      |//////\    |//////\           /\___/\
            |//| )//|   |//| |//|         |       |
            |//////<    |//| |//|        _  *   *  _
            |//| )//|   |//|_|//|        -   /_\   -
            |/////// #  |/////// #           ---
```

Email: bdonal@elaine.ee.und.ac.za
Fax: (27) 31-8162111

Date: 3 Aug 93 14:30:26 GMT
From: ogicse!emory!wa4mei!ke4zv!gary@network.ucsd.edu
Subject: Single frequency receiver
To: ham-homebrew@ucsd.edu

In article <1925@arrl.org> zlau@arrl.org (Zack Lau) writes:
>
> Yes, amplifiers can certainly become oscillators. However, many
>amateur stations generate a -10 dBm signal that becomes a +60 dBm signal,
>all on the same frequency! Thus, it certainly is possible to get 70
>dB of gain on one frequency. Since tuned radio frequency receivers use
>two frequency bands (audio and RF), getting 140 dB of gain out of a TRF
>is certainly feasible. By going to 3 frequency bands, (audio, IF, and
>RF), you can have a receiver that has very little shielding--just a good
>layout. This is quite important to a manufacturer who wants to make
>millions of cheap receivers at a competitive price.

Or amateurs with less than optimum construction skills. It's certainly
easier to tame a system that starts with internally generated drive at
-10 dbm than one that starts with external signals at -120 dbm for most
constructors. Keeping the feedback leakage that low can be a challenge.
It's certainly not impossible, as TRF receivers of the past demonstrate.

> Finally, there is a good reason for not using mixers if at all possible.
>They are typically the weak link in the system--they generate distortion
>products that are often objectionable. Mechanical filter and crystals
>can also cause problems, particularly when hit with too much signal. Thus,
>direct conversion and TRF systems often sound much better that more
>conventional commercial approaches.

They can indeed. I think that taming these beasts can be much more
of a challenge than with a superhet, however. Many DC receivers,
where all the gain is at audio frequencies, have stability problems,
and hum and noise pickup problems, not to mention image problems.
Still, some perform remarkably well. And old TRF receivers were
incredibly touchy to tune, easily breaking into oscillation with
the wrong combination of settings. Even the wind blowing the antenna
would often trigger them into oscillation. A TRF built for a single
frequency might perform well if constructed well, but it wouldn't be
my first choice due to the numerous chances for problems.

I've listened to old TRF radios and been amazed at their clarity compared
to superhets of the same vintage, but they don't have AGC, and the tubes
of the day were more non-linear than any of the circuits used in modern
superhets, and the shape factor of their LC networks doesn't compare to
that of modern crystal or mechanical filters. All but the latter might be
alleivated in a modern design. Perhaps even the shape factor could be
improved with proper stagger tuning on a fixed frequency receiver. It

might be an interesting challenge to build a modern TRF, but if I wanted
a practical receiver, I'd likely stay with a superhet.

Gary
--
Gary Coffman KE4ZV | You make it, | gatech!wa4mei!ke4zv!gary
Destructive Testing Systems | we break it. | uunet!rsiatl!ke4zv!gary
534 Shannon Way | Guaranteed! | emory!kd4nc!ke4zv!gary
Lawrenceville, GA 30244 | |

Date: Tue, 3 Aug 1993 14:41:43 GMT
From: dog.ee.lbl.gov!overload.lbl.gov!agate!usenet.ins.cwru.edu!gatech!wa4mei!
ke4zv!gary@network.ucsd.edu
Subject: Single frequency receiver
To: ham-homebrew@ucsd.edu

In article <CB5v8t.925@srgenprp.sr.hp.com> alanb@sr.hp.com (Alan Bloom) writes:
>Michael Covington (mcovingt@aisun3.ai.uga.edu) wrote:
>
>: What kind of detector would you use in a high-fidelity single-channel
>: AM receiver? Synchronous or conventional?
>
>Synchronous detectors have much lower distortion than diode detectors.
>They also reduce the distortion caused by selective fading. However, they
>are more complicated and harder to get working right.

However, they have a threshold. For DXing a weak AM station, a diode
detector, or using pseudo-sync detection of a single sideband will give
less distortion. If the signal is moderately strong, the sync detector
wins hands down.

The reason for the threshold is that as the signal becomes weaker,
the sync lock becomes noisy due to disturbances from random channel
noise. This causes phase jitter that creates distortion. Because of
the way a sync detector works, this distortion is 2X that of a diode
detector under the same conditions.

Gary

--
Gary Coffman KE4ZV | You make it, | gatech!wa4mei!ke4zv!gary
Destructive Testing Systems | we break it. | uunet!rsiatl!ke4zv!gary
534 Shannon Way | Guaranteed! | emory!kd4nc!ke4zv!gary
Lawrenceville, GA 30244 | |

Date: Tue, 3 Aug 1993 14:51:06 GMT
From: dog.ee.lbl.gov!overload.lbl.gov!agate!usenet.ins.cwru.edu!gatech!wa4mei!ke4zv!gary@network.ucsd.edu
To: ham-homebrew@ucsd.edu

References <25230@acorn.co.uk>, <1993Jul30.141531.16626@ke4zv.uucp>, <25277@acorn.co.uk>
Reply-To : gary@ke4zv.UUCP (Gary Coffman)
Subject : Re: What equipment to tune up a Ramsey 146?

In article <25277@acorn.co.uk> agodwin@acorn.co.uk (Adrian Godwin) writes:
>In article <1993Jul30.141531.16626@ke4zv.uucp> gary@ke4zv.UUCP (Gary Coffman) writes:
>>
>>I use them mainly to check component gates when troubleshooting
>>equipment that *once* worked. Trying to debug a design is a different
>>problem. Ideally, I'd have a current probe instead of a logic probe.
>>HP used to make one, but I haven't found one lately, and HP wanted
>>too much money. The current probe is a great aid when you have a
>>hammered line. It can tell you if the problem is a shorted output
>>or input without having to cut traces.
>>
>
>I'd like to play with one of these - am I right in thinking they use a
>magnetic, rather than a voltage pickup ? If so, there might be a problem
>in finding the right signal on modern boards with closely spaced tracks.
>However, it looks like an interesting homebrew project - maybe using
>a pickup based on a junked floppy disc drive head ?

Yes, they are pickups dependent on the magnetic field generated by
the current flow. A floppy head might work fine for *pulses*, but
I think Hall effect sensors are used in commercial probes. Since
you want to read DC currents as well as AC, an inductive pickup
won't work.

Gary
--
```
Gary Coffman KE4ZV          |    You make it,     | gatech!wa4mei!ke4zv!gary
Destructive Testing Systems |    we break it.     | uunet!rsiatl!ke4zv!gary
534 Shannon Way             |    Guaranteed!      | emory!kd4nc!ke4zv!gary
Lawrenceville, GA 30244     |                     |
```

Date: 3 Aug 1993 15:33:12 GMT
From: noc.near.net!bigboote.WPI.EDU!duck!jmhill@uunet.uu.net
To: ham-homebrew@ucsd.edu

References <careyj.743982155@spot.Colorado.EDU>, <CAyJpL.473@ssesco.com>, <23c2dq$evg@apple.com>
Subject : Re: How do TDR's work

I guess I missed the original note, sorry;

TEKTRONIXS has a neat TDR that is basically a stand alone unit, it even plots out a graph of the cable characteristic, its an analog unit. The unit has a set of switches for selecting the correct cable impedance, and a set of switches for selecting the velecity factor. Unfortuenately it seems that ANY nice lab gear is out of the budget of most hams.

The basic idea is simple, transmit a pulse into the cable that is relatively short in comparison to the time it takes for the signal to take a round trip down the cable and back. At the same time monitor the voltage at the same end of the cable. Any discontinuties in the cable cause refelctions that you will, "see" while you monitor the cable end.

If you have an oscilloscope and a function generator that are both fairly fast, try it yourself. Get your hand calculator out so that you can convert the time you see on the scope to distance. If you start with a known length of cable, you can calculate the velocity factor. Make sure to terminate the end of the cable you are working with, 50 ohms for RG58 or RG8, you don't want to re-reflect the signals that come back to you.

Jonathan KA1WZN

End of Ham-Homebrew Digest V93 #1
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